PATENT COOPERATION TREATY

From the INTERNATIONAL BUREAU

PCT

NOTIFICATION OF THE RECORDING OF A CHANGE

(PCT Rule 92bis.1 and Administrative Instructions, Section 422)

To:

DE BEAUMONT, Michel
Cabinet Michel de Beaumont
1, rue Champollion
F-38000 Grenoble
FRANCE

Date of mailing (day/month/year) 21 August 2000 (21.08.00)	
Applicant's or agent's file reference Case 679 PCT	**IMPORTANT NOTIFICATION**
International application No. PCT/SE99/00790	**International filing date** (day/month/year) 11 May 1999 (11.05.99)

1. The following indications appeared on record concerning:

☐ the applicant ☐ the inventor ☒ the agent ☐ the common representative

Name and Address	State of Nationality	State of Residence
PRAGSTEN, Rolf Telia Research AB Vitsandsgatan 9 S-123 86 Farsta Sweden	Telephone No. 46 8 713 83 27 Facsimile No. 46 8 713 83 21 Teleprinter No.	

2. The International Bureau hereby notifies the applicant that the following change has been recorded concerning:

☒ the person ☐ the name ☐ the address ☐ the nationality ☐ the residence

Name and Address	State of Nationality	State of Residence
DE BEAUMONT, Michel Cabinet Michel de Beaumont 1, rue Champollion F-38000 Grenoble France	Telephone No. 476 51 84 51 Facsimile No. 476 44 62 54 Teleprinter No.	

3. Further observations, if necessary:

4. A copy of this notification has been sent to:

☒ the receiving Office
☐ the International Searching Authority
☒ the International Preliminary Examining Authority
☐ the designated Offices concerned
☒ the elected Offices concerned
☐ other:

The International Bureau of WIPO **34, chemin des Colombettes** **1211 Geneva 20, Switzerland** Facsimile No.: (41-22) 740.14.35	Authorized officer S. De Michiel Telephone No.: (41-22) 338.83.38

PATENT COOPERATION TREATY

PCT

NOTIFICATION OF THE RECORDING OF A CHANGE

(PCT Rule 92bis.1 and Administrative Instructions, Section 422)

From the INTERNATIONAL BUREAU

To:

DE BEAUMONT, Michel
Cabinet Michel de Beaumont
1, rue Champollion
F-38000 Grenoble
FRANCE

Date of mailing (day/month/year) 21 August 2000 (21.08.00)	
Applicant's or agent's file reference Case 679 PCT	**IMPORTANT NOTIFICATION**
International application No. PCT/SE99/00790	**International filing date** (day/month/year) 11 May 1999 (11.05.99)

1. The following indications appeared on record concerning:

[X] the applicant [ ] the inventor [ ] the agent [ ] the common representative

Name and Address	State of Nationality	State of Residence
TELIA AB (publ) Mårbackagatan 11 S-123 86 Farsta Sweden	SE	SE
	Telephone No.	
	Facsimile No.	
	Teleprinter No.	

2. The International Bureau hereby notifies the applicant that the following change has been recorded concerning:

[X] the person [ ] the name [ ] the address [ ] the nationality [ ] the residence

Name and Address	State of Nationality	State of Residence
STMICROELECTRONICS NV World Trade Center Strawinskylaan 1725 NL-1077 XX Amsterdam Netherlands	NL	NL
	Telephone No.	
	Facsimile No.	
	Teleprinter No.	

3. Further observations, if necessary:

4. A copy of this notification has been sent to:

[X] the receiving Office
[ ] the International Searching Authority
[X] the International Preliminary Examining Authority
[ ] the designated Offices concerned
[X] the elected Offices concerned
[ ] other:

The International Bureau of WIPO **34, chemin des Colombettes** **1211 Geneva 20, Switzerland** Facsimile No.: (41-22) 740.14.35	Authorized officer S. De Michiel Telephone No.: (41-22) 338.83.38

PCT

WORLD INTELLECTUAL PROPERTY ORGANIZATION
International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : H04L 5/06, 27/26 — A1

(11) International Publication Number: WO 99/60743

(43) International Publication Date: 25 November 1999 (25.11.99)

(21) International Application Number: PCT/SE99/00790

(22) International Filing Date: 11 May 1999 (11.05.99)

(30) Priority Data:
9801748-6 18 May 1998 (18.05.98) SE

(71) Applicant (*for all designated States except US*): TELIA AB (publ) [SE/SE]; Mårbackagatan 11, S-123 86 Farsta (SE).

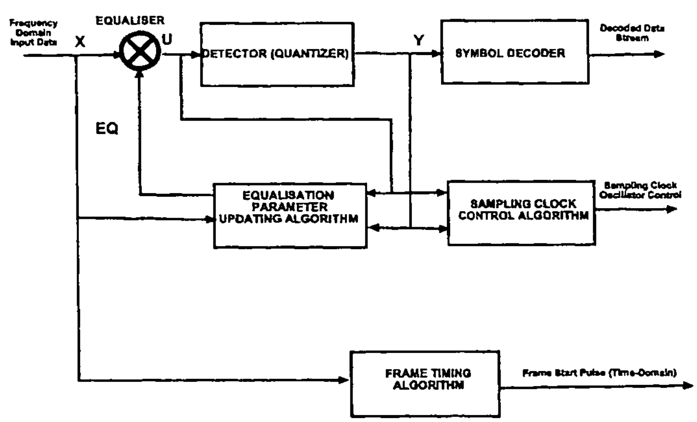
(72) Inventors; and
(75) Inventors/Applicants (*for US only*): JOHANSSON, Magnus [SE/SE]; Timmermansgatan 34, S-972 41 Luleå (SE). OLSSON, Lennart [SE/SE]; Majvägen 39, S-973 31 Luleå (SE). BAHLENBERG, Gunnar [SE/SE]; Blidvägen 234, S-976 32 Luleå (SE). BENGTSSON, Daniel [SE/SE]; Forskarvägen 36 A, S-977 53 Luleå (SE). ISAKSSON, Mikael [SE/SE]; Borgmästarevägen 7, S-973 42 Luleä (SE). OLOFSSON, Sven-Rune [SE/SE]; Malmuddsvägen 9, S-972 45 Luleå (SE). ÖKVIST, Sven, Göran [SE/SE]; Hagaplan 7, S-974 41 Luleå (SE).

(74) Agent: PRAGSTEN, Rolf; Telia Research AB, Vitsandsgatan 9, S-123 86 Farsta (SE).

(81) Designated States: EE, JP, LT, LV, NO, US, European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

Published
With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: IMPROVEMENTS IN, OR RELATING TO, TELECOMMUNICATIONS TRANSMISSION SYSTEMS



(57) Abstract

In a multi-carrier system employing OFDM, for example DMT, an adaptive channel equalizer is normally used, operating in the frequency domain. The internal parameters on which such equalizers operate information that defines a time delay between the transmitter and receiver sampling clock. The sampling clock is controlled so that the time delay between the transmitter and the receiver is effectively eliminated. If the information used to control the sampling clock is received from the equalized data stream, it will introduce an ambiguity between the operation of the channel equalizer and the mechanism used to control the sampling clock. Operation of the equalizer can mask an increasing time difference, between transmitter and receiver, which the sample clock controller should be tracking. The present invention eliminates the ambiguities in the operation of the equalizer and sample clock controller by preventing the equalizer accepting time differences which should be corrected by operation of the sample clock controller. The method of the present invention incorporates a modified algorithm for updating the equalizer's parameters. The present invention may be used in, for example, ADSL and VDSL systems employing DMT which have relatively stationary channels. The principle, however, has general application and may be used, with advantage, in mobile and semi-mobile systems such as DECT and GSM.

FOR THE PURPOSES OF INFORMATION ONLY

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

AL	Albania
AM	Armenia
AT	Austria
AU	Australia
AZ	Azerbaijan
BA	Bosnia and Herzegovina
BB	Barbados
BE	Belgium
BF	Burkina Faso
BG	Bulgaria
BJ	Benin
BR	Brazil
BY	Belarus
CA	Canada
CF	Central African Republic
CG	Congo
CH	Switzerland
CI	Côte d'Ivoire
CM	Cameroon
CN	China
CU	Cuba
CZ	Czech Republic
DE	Germany
DK	Denmark
EE	Estonia
ES	Spain
FI	Finland
FR	France
GA	Gabon
GB	United Kingdom
GE	Georgia
GH	Ghana
GN	Guinea
GR	Greece
HU	Hungary
IE	Ireland
IL	Israel
IS	Iceland
IT	Italy
JP	Japan
KE	Kenya
KG	Kyrgyzstan
KP	Democratic People's Republic of Korea
KR	Republic of Korea
KZ	Kazakstan
LC	Saint Lucia
LI	Liechtenstein
LK	Sri Lanka
LR	Liberia
LS	Lesotho
LT	Lithuania
LU	Luxembourg
LV	Latvia
MC	Monaco
MD	Republic of Moldova
MG	Madagascar
MK	The former Yugoslav Republic of Macedonia
ML	Mali
MN	Mongolia
MR	Mauritania
MW	Malawi
MX	Mexico
NE	Niger
NL	Netherlands
NO	Norway
NZ	New Zealand
PL	Poland
PT	Portugal
RO	Romania
RU	Russian Federation
SD	Sudan
SE	Sweden
SG	Singapore
SI	Slovenia
SK	Slovakia
SN	Senegal
SZ	Swaziland
TD	Chad
TG	Togo
TJ	Tajikistan
TM	Turkmenistan
TR	Turkey
TT	Trinidad and Tobago
UA	Ukraine
UG	Uganda
US	United States of America
UZ	Uzbekistan
VN	Viet Nam
YU	Yugoslavia
ZW	Zimbabwe

Improvements in, or Relating to, Telecommunications Transmission Systems

The present invention relates to multi-carrier telecommunications transmission systems employing OFDM (Orthogonal Frequency Division Multiplexing), receivers for us in multi-carrier telecommunications transmission systems, and methods of maintaining sampling clock synchronisation between a transmitter sampling clock and a receiver sampling clock in multi-carrier telecommunications transmission systems.

In a multi-carrier system of the OFDM type, including DMT (Discrete Multi-Tone), the control of a receiver's sampling clock and the updating of the equalizer's parameters may, in some cases, interfere. The present invention coordinates the equalizer's function and control of the sampling clock, in a new way, such that this problem is eliminated.

In a multi-carrier system employing OFDM, for example DMT, an adaptive channel equalizer is normally used, operating in the frequency domain. The internal parameters on which such equalizers operate include, in addition to data relating to the characteristics of the channel, information that defines a time delay between the transmitter and receiver sampling clock. The sampling clock is controlled so that the time delay between the transmitter and the receiver is effectively eliminated. If the information used to control the sampling clock is received from the equalized data stream, it will introduce an ambiguity between the operation of the channel equalizer and the mechanism used to control the sampling clock. In particular, it will not be clear to what extent the time delays between the transmitter and receiver have been eliminated by the sampling clock and/or by the equalizer. Operation of the equalizer can mask an increasing time difference, between transmitter and receiver, which the sample clock controller should be tracking. This may lead to a situation where the sampling of one frame, in the time domain, occurs at a time which is so long after the correct time period that inter symbol interference occurs. The present invention eliminates the ambiguities in the operation of the equalizer and sample clock controller by preventing the equalizer accepting time differences which should be corrected by operation of the sample clock controller. The method of the present invention incorporates a modified

algorithm for updating the equalizer's parameters.

The present invention may be used in, for example, ADSL and VDSL systems employing DMT which have relatively stationary channels. The principle, however, has general application and may be used, with advantage, in mobile and semi-mobile systems such as DECT and GSM.

According to a first aspect of the present invention, there is provided a receiver, for use in an OFDM transmission system, having an adaptive channel equalizer means, a sampling clock and a sampling clock control means characterised in that ambiguity prevention means are provided to prevent said adaptive channel equalizer means from operating on time differences which should be corrected by operation of said sampling clock control means.

Said sampling clock may be controlled by data derived from an equalized data stream.

Sampling time deviations in said OFDM system may cause received frame argument functions to have a linear slope and said sampling clock may be controlled using an estimate of said frame argument functions' slope.

Said adaptive channel equalizer may be prevented from operating on said time differences by forcing the slope of a linear part of an equalizer parameter argument function to be always zero.

Said sampling clock's frequency may be controlled by a feed-back signal generated from an estimated slope of an argument function, $Y^*.U$ which is the element-by-element product of an equalizer output vector U and the conjugate of a quantized vector Y derived from an output of a detector means operating on U.

A slope of said equalizer parameter argument function may be derived by taking an average slope of the equalizer parameter argument function by unwrapping said equalizer parameter argument function and deriving said average slope from said unwrapped equalizer parameter argument function.

The average slope α_k of the linear part of the equalizer parameter argument function may be calculated from:

$$\alpha_k = \frac{1}{N}\sum_n \frac{\angle EQ_{n,k}}{n} \qquad (1a)$$

where $\angle$EQ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number and N is the size of the frequency domain frame.

The average slope α_k of the linear part of the equalizer parameter argument function may be calculated from:

$$\alpha_k = \frac{2}{n_2 - n_0}\left(\sum_{n=n_1+1}^{n_2} \angle EQ_{n,k} - \sum_{n=n_0}^{n_1} \angle EQ_{n,k}\right) \qquad (1b)$$

where $\angle$EQ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number, N is the size of the frequency domain frame, n_1 divides the received frequency band into two equal parts and n_0 and n_2 are lower and upper limits, respectively, of the frequency band.

Where several separate frequency bands are present in the received signal, equation 1(b) may be applied to each frequency band separately and the average of the results employed as the slope of the equalizer parameter argument function.

Said equalizer parameter argument function may be rotated in small steps until said slope is zero.

Said rotation may be performed by using a vector L of complex numbers with unit magnitude and a linear argument function with a slope equal to a small fraction α_k, and in that L is calculated from:

$$L_{n,k} = \exp(-j.\beta.\frac{n}{N}.\alpha_k) \qquad (2)$$

where β controls the speed of adaption to the zero slope.

An equaliser parameter vector EQ may be adaptively updated using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_1 . \frac{X^*_{n,k}}{|X_{n,k}|^2} . (Y_{n,k} - U_{n,k})] . L_{n,k} \qquad (3a)$$

An equaliser parameter vector EQ may be adaptively updated using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_2 . \frac{X^*_{n,k}}{|X_{n,k}|} . (Y_{n,k} - U_{n,k})] . L_{n,k} \qquad (3b)$$

An equaliser parameter vector EQ may be adaptively updated using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_3 . X^*_{n,k} . (Y_{n,k} - U_{n,k})] . L_{n,k} \qquad (3c)$$

The algorithm defined by equation 3(a) may be employed during a start up sequence for said receiver.

The algorithm defined by equation 3(c) may be used for tracking slow changes in the adaptive equaliser parameter EQ subsequent to a start up sequence for said receiver.

Said OFDM system may employ DMT.

Said OFDM system may be an ADSL system.

Said OFDM system may be a VDSL system.

Said OFDM system may be a mobile telecommunications system.

According to a second aspect of the present invention, there is provided an OFDM multi-carrier transmission system having at least one transmitter and a plurality of receivers, characterised in that said receivers are receivers as set forth in any preceding paragraph.

According to a third aspect of the present invention, there is provided a transceiver, for use in an OFDM transmission system, characterised in that said transceiver includes a receiver as set forth in any preceding paragraph.

According to a fourth aspect of the present invention, there is provided, in an OFDM transmission system having a transmitter and a receiver, said receiver having an adaptive channel equalizer means, a sampling clock and a sampling clock control means, and said transmitter having a sampling clock, a method of maintaining synchronisation between said receiver sampling clock and said transmitter sampling clock, characterised by preventing said adaptive channel equalizer means from operating on time differences which should be corrected by operation of said sampling clock control means.

Said sampling clock may be controlled by data derived from an equalized data stream.

Sampling time deviations in said OFDM system may cause received frame argument functions to have a linear slope and said sampling clock may be controlled using an estimate of said frame argument functions' slope.

Said adaptive channel equalizer may be prevented from operating on said time differences by forcing the slope of a linear part of an equalizer parameter argument function to be always zero.

Said sampling clock's frequency may be controlled by a feed-back signal generated from an estimated slope of an argument function, $Y^*.U$ which is the element-by-element product of an equalizer output vector U and the conjugate of a quantized vector Y derived from an output of a detector means operating on U.

A slope of said equalizer parameter argument function may be derived by taking an average slope of the equalizer parameter argument function by unwrapping said equalizer parameter argument function and deriving said average slope from said unwrapped equalizer parameter argument function.

The average slope α_k of the linear part of the equalizer parameter argument function may be calculated from:

$$\alpha_k = \frac{1}{N}\sum_n \frac{\angle EQ_{n,k}}{n} \qquad (1a)$$

where $\angle$EQ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number and N is the size of the frequency domain frame.

The average slope α_k of the linear part of the equalizer parameter argument function may be calculated from:

$$\alpha_k = \frac{2}{n_2 - n_0}\left(\sum_{n=n_1+1}^{n_2} \angle EQ_{n,k} - \sum_{n=n_0}^{n_1} \angle EQ_{n,k}\right) \qquad (1b)$$

where $\angle$EQ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number, N is the size of the frequency domain frame, n_1 divides the received frequency band into two equal parts and n_0 and n_2 are lower and upper limits, respectively, of the frequency band.

Where several separate frequency bands are present in the received signal, equation 1(b) may be applied to each frequency band separately and the average of the results employed as the slope of the equalizer parameter argument function.

Said equalizer parameter argument function may be rotated in small steps until said slope is zero.

Said rotation may be performed by using a vector L of complex numbers with unit magnitude and a linear argument function with a slope equal to a small

fraction α_k, and in that L is calculated from:

$$L_{n,k}=\exp(-j.\beta.\frac{n}{N}.\alpha_k) \qquad (2)$$

where β controls the speed of adaption to the zero slope.

An equaliser parameter vector EQ may be adaptively updated using an algorithm defined by:

$$EQ_{n,k+1}=[EQ_{n,k}+\mu_1.\frac{X^*_{n,k}}{|X_{n,k}|^2}.(Y_{n,k}-U_{n,k})].L_{n,k} \qquad (3a)$$

An equaliser parameter vector EQ may be adaptively updated using an algorithm defined by:

$$EQ_{n,k+1}=[EQ_{n,k}+\mu_2.\frac{X^*_{n,k}}{|X_{n,k}|}.(Y_{n,k}-U_{n,k})].L_{n,k} \qquad (3b)$$

An equaliser parameter vector EQ may be adaptively updated using an algorithm defined by:

$$EQ_{n,k+1}=[EQ_{n,k}+\mu_3.X^*_{n,k}.(Y_{n,k}-U_{n,k})].L_{n,k} \qquad (3c)$$

The algorithm defined by equation 3(a) may be employed during a start up sequence for said receiver.

The algorithm defined by equation 3(c) may be used for tracking slow changes in the adaptive equaliser parameter EQ subsequent to a start up sequence for said receiver.

Said OFDM system may employ DMT.

Said OFDM system may be an ADSL system.

Said OFDM system may be a VDSL system.

Said OFDM system may be a mobile telecommunications system.

Embodiments of the invention will now be described, by way of example, with reference to the accompanying drawings, in which:

Figure 1 illustrates, in block schematic form, the equalizer and sampling clock control unit of a receiver according to the present invention.

Figure 2 illustrates the time domain data format for OFDM.

In a multi-carrier system employing OFDM, for example DMT, an adaptive channel equalizer is normally used, operating in the frequency domain. The equalizer operates on, inter alia, parameters which define the time delay between the transmitter and receiver sampling clocks. The sampling clock, however, must be controlled so that the time delay between the transmitter and the receiver is effectively eliminated. If the information used to control the sampling clock is received from the equalized data stream, there will be an ambiguity between the operation of the channel equalizer and the sampling clock control mechanism. In particular, it will not be clear to what extent time delays between the transmitter and receiver have been eliminated by operation of the sampling clock control mechanism, or the equaliser. Operation of the equalizer can thus mask an increasing time difference, between transmitter and receiver, which the sample clock controller should track. This can lead to a situation where the sampling of one frame, in the time domain, occurs at a time, which is so long after the correct time period, that inter symbol interference occurs. The present invention eliminates the ambiguities in the operation of the equaliser and sample clock controller by preventing the equalizer accepting time differences which should be corrected by operation of the sample clock controller.

A sampling time deviation in an OFDM system causes the received frame argument function to have a linear slope. The sampling clock oscillator is.

therefore, controlled using an estimate of this slope. The estimation of the argument function is performed using equalized data. Since the equalizer adapts to the channel properties, it also eliminates any sampling time deviations. This means that the equalizer and the sampling clock control may mutually cancel each other out, in respect of sampling time deviations. Since the frame timing is set during the start-up sequence, it also relies on these control actions. It is, therefore, necessary to prevent such mutual cancellation.

The method of the present invention solves the mutual cancellation problem by preventing the equalizer from operating on the time delay. This means that the slope of the linear part of the equalizer parameter argument function is always substantially zero. An adaptive parameter estimation algorithm that fulfils this requirement is incorporated in the present invention.

Figure 1 illustrates an OFDM receiver, according to the present invention. Incoming frequency domain data, X, passes to an equalizer and then, as an equalized signal, to a detector, i.e. a quantizer. The output from the detector is passed to a symbol detector from which a decoded data stream emerges. Frame start pulses, in the time domain, are derived from a frame timing algorithm which operates on the received frequency domain input data and runs in a frame timing mechanism. The equalizer is controlled by an equalisation parameter updating algorithm which operates on the frequency domain input data and an output derived from the sampling clock control algorithm. The equaliser is controlled by signal EQ. The sampling clock control algorithm, which operates in the sampling clock control mechanism, produces a signal for controlling the sampling clock oscillator.

The frequency-domain data is the Fourier transform of the received time-domain OFDM frames. The time-domain frames are sampled in synchronism with the transmitter so that each received frame contains data from only one transmitted frame. This is important in order to maintain the orthogonality of the frames.

A common time-domain format for the transmission of OFDM frames, is illustrated in Figure 2, where CE = Cyclic Extension.

The signalling interval contains a cyclic extension and a frame. The cyclic

extension is a copy of the opposite part of the frame. This means that a frame sampled anywhere inside the signalling interval will contain data from one transmitted frame only, in the correct order. A deviation from the exact frame timing leads to a cyclic permutation of the frame, but the orthogonality is still maintained. The cyclic extension may be split into two parts, namely a prefix part and a suffix part.

A training procedure is required at start-up. The frame timing is adjusted until the received frames are sampled within the frame interval. The sampling clock frequency must also be adjusted so that it is sufficiently close to the transmitter clock frequency to enable the equalizer to estimate a fairly stable set of parameters.

The generation of frame start pulses is achieved by counting sampling clock intervals. Therefore, after the initial setting of the sampling clock timing during the training procedure, maintenance of synchronisation between a receiver and a transmitter will be dependent on the feed-back control of the sampling clock oscillator. The clock oscillator frequency is controlled by a feed-back signal generated from the estimated slope of the argument function of vector Y^*U; which is the element-by-element product of the conjugate of the quantized vector Y and the equalizer output vector U.

After the training procedure the equalizer parameters EQ will represent the complex frequency-domain inverse of the channel. The product vector Y^*U will then have a linear argument function with a slope representing the time deviation between the transmitter and receiver sample timing. However, the argument function of EQ may also have a linear part with non-zero slope unless measures are taken to prevent this occurring. This also corresponds to a deviation between the transmitter and receiver sample timing. Since the vector EQ represents the inverse channel, the slope of EQ has the opposite sign to the slope of Y^*U for the same time deviation. Therefore, they may become large and still cancel each other out. This is a potential problem, since the frame timing can gradually drift away from the frame interval and inter-symbol interference will eventually increase.

The solution to this problem, proposed by the present invention, is to

prevent the equalizer parameter vector EQ from representing time delay. This means that the linear part of its argument function must be forced to zero slope. The sample time deviation will then be represented by the argument function slope of $Y^{*}U$ and taken care of by the sampling clock oscillator feed-back controller.

The equalizer parameter estimation is performed adaptively using, as input data, the frequency-domain vectors X, U and Y, see Figure 1. During the first part of the start-up sequence, the vector Y is temporarily replaced by a training frame T with known content. In the equations, set out in this patent specification, k is the frame number, n is the carrier index and N is the size of the frequency-domain frame.

The novel feature of the algorithm employed by the present invention is that any non-zero time delay representation included in the equalizer parameter vector EQ will be eliminated.

The exact linear argument function representing the time delay is not available, but an approximation can be estimated. The argument function of the equalizer parameter vector is generally non-linear, but a linear part can be found by taking the average slope of the argument function. This means that the unwrapped argument function has to be calculated and the average slope of this used as a measure of the time delay. Thus, the algorithm will use this as a feed-back value to successively rotate the argument function in small steps until the average slope is zero.

The argument function of the equalizer parameters is the vector of arguments of the individual complex elements. The argument of a complex number is the inverse tangent of the imaginary part divided by the real part. A problem involved in this calculation is that the inverse tangent function is periodic with a period 2π radians. In this application it is necessary to handle larger arguments than π radians, which is the range of the inverse tangent function. An assumption used here is that the difference in argument between adjacent parameters is smaller than π radians. It is then possible to compensate for each discontinuity due to the inverse tangent function periodicity and thus unwrap the argument function.

The average slope α_k of the linear part can be calculated as shown in Equation (1a), or by some other standard method, using the unwrapped argument function of EQ.

$$\alpha_k = \frac{1}{N}\sum_n \frac{\angle EQ_{n,k}}{n} \qquad (1a)$$

If the lowest frequency carriers are not present in the frame, it is not possible to find the true argument function since there will be an unknown starting value for the available part of the function. This is not a problem in the present application, since the average slope can still be calculated.

Equation (1b) shows an alternative algorithm that gives the average slope of a contiguous band of active carriers.

$$\alpha_k = \frac{2}{n_2 - n_0}\left(\sum_{n=n_1+1}^{n_2} \angle EQ_{n,k} - \sum_{n=n_0}^{n_1} \angle EQ_{n,k}\right) \qquad (1b)$$

Indexes n_0 and n_2 are the lower and upper limits respectively of the band. Index n_1 divides the band into two equal parts. If several separate bands are used, equation (1b) is applied to each band and the average of the results is calculated. The algorithm according to equation (1b) gives a very simple hardware implementation.

The rotation operation is performed using a vector L of complex numbers with unit magnitude and a linear argument function with a slope equal to a small fraction of α_k. The vector L is calculated using Equation (2).

$$L_{n,k} = \exp(-j.\beta.\frac{n}{N}.\alpha_k) \qquad (2)$$

The coefficient β controls the speed of adaption to the zero average slope of the *EQ* vector argument function.

Different algorithms for the adaptive updating of the equalizer parameter vector *EQ* are given by Equations (3a), (3b) and (3c). They all include the new feature to eliminate the time delay representation. One specific feature that characterises the present invention is the multiplication by the rotational vector *L*.

The algorithm according to Equation (3a) is the fastest approach, while Equation (3c), which is a modified LMS algorithm, has the best performance for low SNR. The selection is not obvious, but Equation (3a) can be used during part of the start-up sequence and Equation (3c) can be used for the tracking of subsequent slow changes.

$$EQ_{n,k+1}=[EQ_{n,k}+\mu_1\cdot\frac{X^*_{n,k}}{|X_{n,k}|^2}.(Y_{n,k}-U_{n,k})].L_{n,k} \qquad (3a)$$

$$EQ_{n,k+1}=[EQ_{n,k}+\mu_2\cdot\frac{X^*_{n,k}}{|X_{n,k}|}.(Y_{n,k}-U_{n,k})].L_{n,k} \qquad (3b)$$

$$EQ_{n,k+1}=[EQ_{n,k}+\mu_3.X^*_{n,k}.(Y_{n,k}-U_{n,k})].L_{n,k} \qquad (3c)$$

The main innovation of the present invention is the principle that the equalizer should be prevented from representing time delay.

The present invention can be realised as a method of implementing this principle as a part of the equalizer parameter updating algorithm.

The sample timing control using this method is very robust in the case of disturbances, since every active carrier is used in the timing deviation estimation.

CLAIMS

1. A receiver, for use in an OFDM transmission system, having an adaptive channel equalizer means, a sampling clock and a sampling clock control means characterised in that ambiguity prevention means are provided to prevent said adaptive channel equalizer means from operating on time differences which should be corrected by operation of said sampling clock control means.

2. A receiver, as claimed in claim 1, characterised in that said sampling clock is controlled by data derived from an equalized data stream.

3. A receiver, as claimed in either claim 1, or claim 2, characterised in that sampling time deviations in said OFDM system cause received frame argument functions to have a linear slope and in that said sampling clock is controlled using an estimate of said frame argument functions' slope.

4. A receiver, as claimed in any previous claim, characterised in that said adaptive channel equalizer is prevented from operating on said time differences by forcing the slope of a linear part of an equalizer parameter argument function to be always zero.

5. A receiver, as claimed in claim 4, characterised in that said sampling clock's frequency is controlled by a feed-back signal generated from an estimated slope of an argument function, $Y^*.U$ which is the element-by-element product of an equalizer output vector U and the conjugate of a quantized vector Y derived from an output of a detector means operating on U.

6. A receiver, as claimed in either claim 4, or 5, characterised in that a slope of said equalizer parameter argument function is derived by taking an average slope of the equalizer parameter argument function by unwrapping said equalizer parameter argument function and deriving said average slope from said unwrapped equalizer parameter argument function.

7. A receiver, as claimed in claim 6, characterised in that the average slope

α_k of the linear part of the equalizer parameter argument function is calculated from:

$$\alpha_k = \frac{1}{N}\sum_n \frac{\angle EQ_{n,k}}{n} \qquad (1a)$$

where $\angle$EQ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number and N is the size of the frequency domain frame.

8. A receiver, as claimed in claim 6, characterised in that the average slope α_k of the linear part of the equalizer parameter argument function is calculated from:

$$\alpha_k = \frac{2}{n_2 - n_0}\left(\sum_{n=n_1+1}^{n_2} \angle EQ_{n,k} - \sum_{n=n_0}^{n_1} \angle EQ_{n,k}\right) \qquad (1b)$$

where $\angle$EQ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number, N is the size of the frequency domain frame, n_1 divides the received frequency band into two equal parts and n_0 and n_2 are lower and upper limits, respectively, of the frequency band.

9. A receiver, as claimed in claim 8, characterised in that, where several separate frequency bands are present in the received signal, equation 1(b) is applied to each frequency band separately and the average of the results employed as the slope of the equalizer parameter argument function.

10. A receiver, as claimed in any of claims 5 to 9, characterised in that said equalizer parameter argument function is rotated in small steps until said slope is zero.

11. A receiver, as claimed in claim 12, characterised in that said rotation is performed by using a vector L of complex numbers with unit magnitude and a linear argument function with a slope equal to a small fraction α_k, and in that L is calculated from:

$$L_{n,k} = \exp(-j.\beta.\frac{n}{N}.\alpha_k) \qquad (2)$$

where β controls the speed of adaption to the zero slope.

12. A receiver, as claimed in any of claims 5 to 11, characterised in that an equaliser parameter vector EQ is adaptively updated using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_1 . \frac{X^*_{n,k}}{|X_{n,k}|^2} .(Y_{n,k} - U_{n,k})].L_{n,k} \qquad (3a)$$

13. A receiver, as claimed in any of claims 5 to 11, characterised in that an equaliser parameter vector EQ is adaptively updated using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_2 . \frac{X^*_{n,k}}{|X_{n,k}|} .(Y_{n,k} - U_{n,k})].L_{n,k} \qquad (3b)$$

14. A receiver, as claimed in any of claims 5 to 11, characterised in that an equaliser parameter vector EQ is adaptively updated using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_3 . X^*_{n,k} .(Y_{n,k} - U_{n,k})].L_{n,k} \qquad (3c)$$

15. A receiver, as claimed in claim 12, characterised in that the algorithm defined by equation 3(a) is employed during a start up sequence for said receiver.

16. A receiver, as claimed in claim 13 and/or 15, characterised in that the algorithm defined by equation 3(c) is used for tracking slow changes in the adaptive equaliser parameter EQ subsequent to a start up sequence for said receiver.

17. A receiver, as claimed in any previous claim, characterised in that said OFDM system employs DMT.

18. A receiver, as claimed in any previous claim, characterised in that said OFDM system is an ADSL system.

19. A receiver, as claimed in any of claims 1 to 17, characterised in that said OFDM system is a VDSL system.

20. A receiver, as claimed in any of claims 1 to 17, characterised in that said OFDM system is a mobile telecommunications system.

21. An OFDM multi-carrier transmission system having at least one transmitter and a plurality of receivers, characterised in that said receivers are receivers as claimed in any of claims 1 to 20.

22. A transceiver, for use in an OFDM transmission system, characterised in that said transceiver includes a receiver as claimed in any of claims 1 to 20.

23. In an OFDM transmission system having a transmitter and a receiver, said receiver having an adaptive channel equalizer means, a sampling clock and a sampling clock control means, and said transmitter having a sampling clock, a method of maintaining synchronisation between said receiver sampling clock and said transmitter sampling clock, characterised by preventing said adaptive channel equalizer means from operating on time differences which should be corrected by operation of said sampling clock control means.

24. A method, as claimed in claim 23, characterised by controlling said sampling clock with data derived from an equalized data stream.

25. A method, as claimed in either claim 23, or claim 24, characterised by sampling time deviations in said OFDM system causing received frame argument functions to have a linear slope and by controlling said sampling clock using an estimate of said frame argument functions' slope.

26. A method, as claimed in any of claims 23 to 25, characterised by preventing said adaptive channel equalizer from operating on said time differences by forcing the slope of a linear part of an equalizer parameter argument function to be always

zero.

27. A method, as claimed in claim 26, characterised by controlling said sampling clock's frequency with a feed-back signal generated from an estimated slope of an argument function, $Y^{*}.U$ which is the element-by-element product of an equalizer output vector U and the conjugate of a quantized vector Y derived from an output of a detector means operating on U.

28. A method, as claimed in either claim 26, or 27, characterised by deriving a slope of said equalizer parameter argument function by taking an average slope of the equalizer parameter argument function by unwrapping said equalizer parameter argument function and deriving said average slope from said unwrapped equalizer parameter argument function.

29. A method, as claimed in claim 28, characterised by calculating the average slope α_k of the linear part of the equalizer parameter argument function from:

$$\alpha_k = \frac{1}{N}\sum_n \frac{\angle EQ_{n,k}}{n} \qquad (1a)$$

where $\angle EQ$ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number and N is the size of the frequency domain frame.

30. A method, as claimed in claim 28, characterised by calculating the average slope α_k of the linear part of the equalizer parameter argument function from:

$$\alpha_k = \frac{2}{n_2 - n_0}\left(\sum_{n=n_1+1}^{n_2} \angle EQ_{n,k} - \sum_{n=n_0}^{n_1} \angle EQ_{n,k}\right) \qquad (1b)$$

where $\angle EQ$ is the unwrapped equalizer parameter argument function, n is the carrier index, k is the frame number, N is the size of the frequency domain frame, n_1 divides the received frequency band into two equal parts and n_0 and n_2 are lower and upper limits, respectively, of the frequency band.

31. A method, as claimed in claim 30, characterised by, where several separate frequency bands are present in the received signal, applying equation 1(b) to each frequency band separately and the average of the results employed as the slope of the equalizer parameter argument function.

32. A method, as claimed in any of claims 27 to 31, characterised by rotating said equalizer parameter argument function in small steps until said slope is zero.

33. A method, as claimed in claim 32, characterised by performing said rotation by using a vector L of complex numbers with unit magnitude and a linear argument function with a slope equal to a small fraction α_k, and in that L is calculated from:

$$L_{n,k} = \exp(-j.\beta.\frac{n}{N}.\alpha_k) \qquad (2)$$

where β controls the speed of adaption to the zero slope.

34. A method, as claimed in any of claims 27 to 33, characterised by adaptively updating an equaliser parameter vector EQ using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_1 . \frac{X^*_{n,k}}{|X_{n,k}|^2} . (Y_{n,k} - U_{n,k})] . L_{n,k} \qquad (3a)$$

35. A method, as claimed in any of claims 27 to 33, characterised by adaptively updating an equaliser parameter vector EQ using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_2 . \frac{X^*_{n,k}}{|X_{n,k}|} . (Y_{n,k} - U_{n,k})] . L_{n,k} \qquad (3b)$$

36. A method, as claimed in any of claims 27 to 33, characterised by adaptively updating an equaliser parameter vector EQ using an algorithm defined by:

$$EQ_{n,k+1} = [EQ_{n,k} + \mu_3 . X^*_{n,k} . (Y_{n,k} - U_{n,k})] . L_{n,k} \qquad (3c)$$

37. A method, as claimed in claim 34, characterised by employing the algorithm defined by equation 3(a) during a start up sequence for said receiver.

38. A method, as claimed in claim 34 and/or 37, characterised by using the algorithm defined by equation 3(c) for tracking slow changes in the adaptive equaliser parameter EQ subsequent to a start up sequence for said receiver.

39. A method, as claimed in any of claims 23 to 38, characterised by said OFDM system employing DMT.

40. A method, as claimed in any of claims 23 to 38, characterised by said OFDM system being an ADSL system.

41. A method, as claimed in any of claims 23 to 38, characterised by said OFDM system being a VDSL system.

42. A method, as claimed in any of claims 23 to 38, characterised by said OFDM system being a mobile telecommunications system.

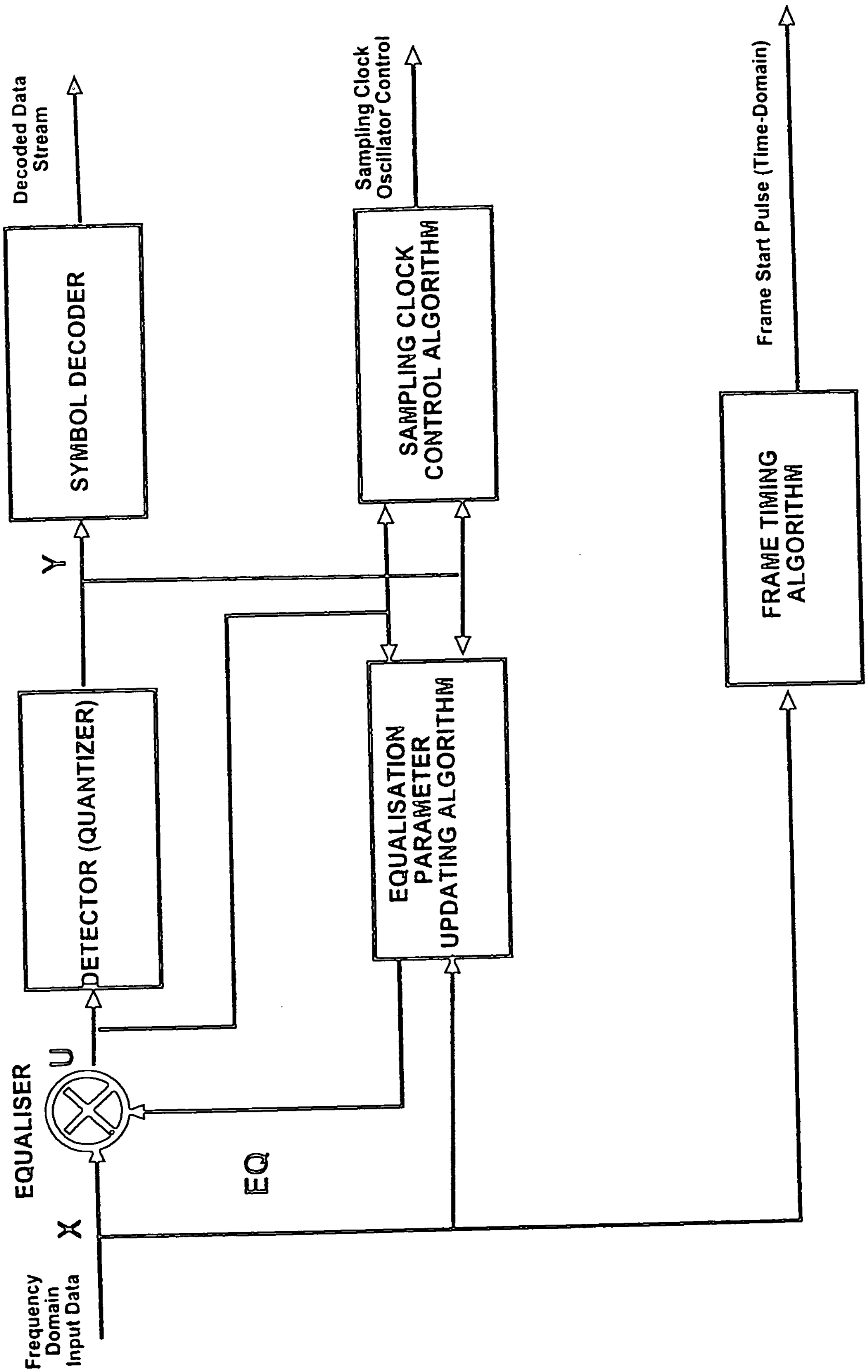
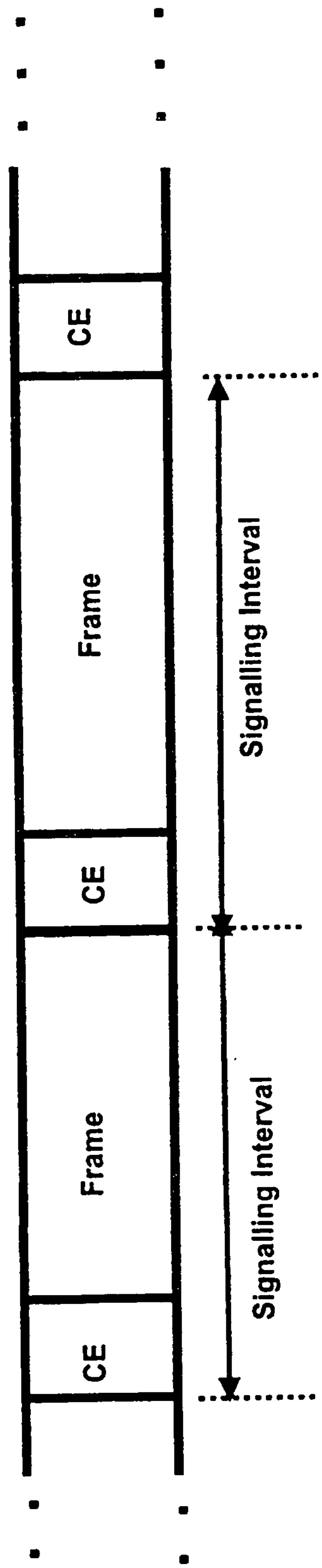

Decoded Data Stream
Sampling Clock Oscillator Control
Frame Start Pulse (Time-Domain)
SYMBOL DECODER
SAMPLING CLOCK CONTROL ALGORITHM
FRAME TIMING ALGORITHM
Y
DETECTOR (QUANTIZER)
EQUALISATION PARAMETER UPDATING ALGORITHM
EQUALISER
U
EQ
X
Frequency Domain Input Data

FIGURE 1



CE = Cyclic Extension

FIGURE 2

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 99/00790

A. CLASSIFICATION OF SUBJECT MATTER

IPC6: H04L 5/06, H04L 27/26
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC6: H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	An implementation of OFDM receiver for digital terrestrial television broadcasting and its technologies Harada, Y et al Broadcasting Convention 1997, International 1997 Pages 337-342 see page 338, left colun line 17, - page 340, left column, line 23 --	1-42
A	WO 9503656 A1 (TELIA AB), 2 February 1995 (02.02.95), page 4, line 33 - page 6, line 12; page 7, line 15 - line 20 --	1-42

[X] Further documents are listed in the continuation of Box C. [X] See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" erlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance: the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance: the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
26 October 1999	03 -11- 1999
Name and mailing address of the ISA/ Swedish Patent Office Box 5055, S-102 42 STOCKHOLM Facsimile No. +46 8 666 02 86	Authorized officer Peder Gjervaldsaeter/MN Telephone No. +46 8 782 25 00

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 99/00790

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	IEEE Communications Magazine, pages 100-109, Volume 33, No 2, February 1995, H Sari et al, "Transmission techniques for digital terrestrial TV broadcasting", see page 102, right column, line 1 - page 103, right column line 2 --	1-42
A	Low-complex frame synchronization in OFDM systems van de Beek et al Universal Personal Communications. 1995, Record. 1995 Fourth IEEE International Conference on 1995 pages 982-986 see abstract -- --------	1-42

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No.
PCT/SE 99/00790

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9503656 A1	02/02/95	EP 0712555 A	22/05/96
		SE 500986 C	17/10/94
		SE 9302453 A	17/10/94
		US 5652772 A	29/07/97